(12) UK Patent Application (19) GB (11) 2 166 320 A

(43) Application published 30 Apr 1986

(21) Application No 8426955

(22) Date of filing 25 Oct 1984

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(51) INT CL[4]
H04M 11/06 H04L 11/20

(52) Domestic classification
H4K TK

(56) Documents cited
None

(58) Field of search
H4K

(54) Packet switching system

(57) In a packet switching system in which both data and speech packets are handled, speech packets, which have to be handled in a real-time way, may be subjected to unacceptable delay if the medium is busy with a data packet when a speech packet is to be sent. This is because a data packet is relatively long, which would produce an unacceptable delay.

To overcome this, if a speech packet arrives when a data packet is partly sent, the sending of the data packet is interrupted, and the speech packet is sent, and the residue of the data packet is sent when the transmission medium is free of speech. This residue is provided with a new header and handled as if it is a new data packet.

The drawing(s) originally filed was (were) informal and the print here reproduced is taken from a later filed formal copy.

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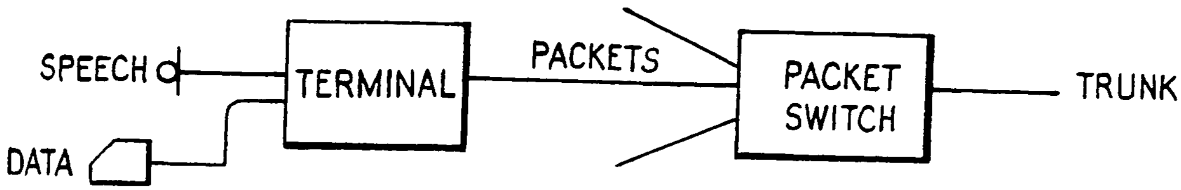

FIG.1

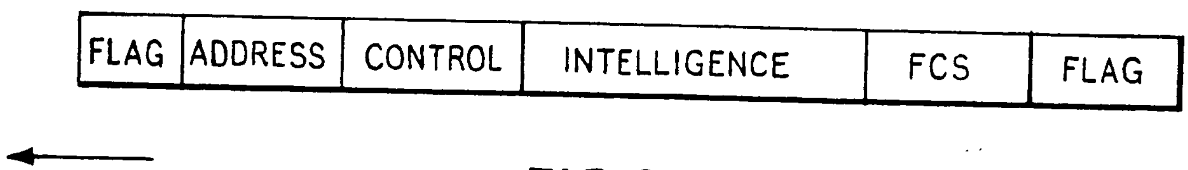

FIG.2

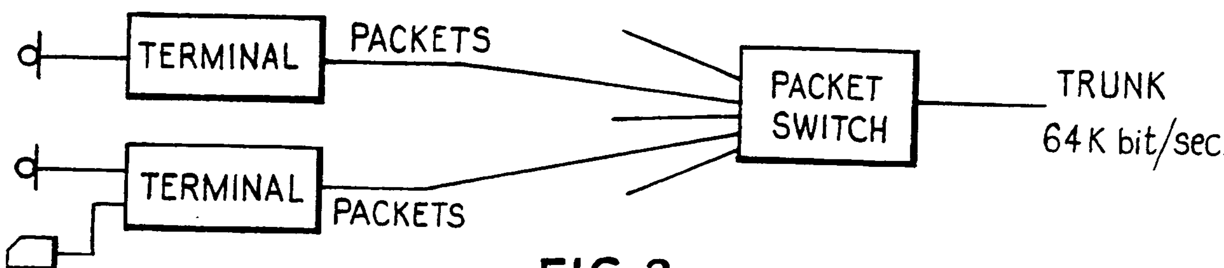

FIG. 3

SPECIFICATION

Packet switching system

This refers to a transmission system in which both data and speech are conveyed in packet form.

In such a system the intelligence to be conveyed is in digital form and is assembled into packets each of which is a block of intelligence with attached to it handling information including a destination address, and message identification. Where a message contains a number of packets they are sent sequentially, but not necessarily over the same route, from the caller to the wanted user. The handling information includes information which ensures that packets are assembled in the correct order.

In such a system, speech, which is essentially a real-time phenomenon, should be delayed as little as possible, whereas in most cases data is tolerant of some delay. An object of the invention is to provide a packet-switched transmission system in which both data and speech are handled without the speech packets being unduly delayed.

According to the present invention, there is provided an electrical intelligence transmission system, in which both speech and data are transmitted as packets over virtual connections set up in a fully digital network, the speech and data packets being transmitted at the bit rate of the transmission medium, in which speech packets have priority over data packets, in which, if a speech packet for transmission is detected during the transmission of a data packet, the transmission of that data packet is interrupted to allow the speech packet to be conveyed over the transmission medium, and in which, when the medium again becomes free from speech, the non-transmitted part of the interrupted data packet is transmitted as a separate data packet.

An embodiment of the invention will now be described with reference to the accompanying drawing, in which

Figure 1 is a simplified explanatory block diagram,

Figure 2 is an intelligence packet of the X25 type, while

Figure 3 is another simplified block diagram.

It is generally accepted that data packets can accept greater delays than speech during transmission, even if the data packets are being handled on an interactive basis. Thus speech should have priority over data at each switching point or node in the network. However, this does not prevent data for which a channel has already been seized from discharging its packet, which may be relatively long.

If, in order to give priority to speech, breakdown of data were involved, this would entail a restart of the data when the speech packet ends, and with a high number of speech users in the network, the delay to data could become important in data periods. In the present system, these difficulties are overcome by artificially ending a data packet being sent when a speech packet has to be sent. The speech packet is then followed by the untransmitted portion of the data packet, preceded by a suitable header, which includes the data packet's address and other handling information.

In assessing the advantages of such a system, it should be understood that the delay to speech due to data packets in a conventional system depends on the speed of transmission of data. Thus if a data packet consists of 1024 bytes, each of eight bits, then at 64K bit/sec, each packet would take 128ms to send. This would mean on average a 64ms. delay to intelligence which follows the packet, which is a substantial delay. If the trunk (or other circuit) involved is a 2Mbit/sec, then the packet would take 4ms, which gives an average delay of 2ms. Hence it will be seen that it is desirable to avoid delay on speech channels. It is very important to avoid such delays on subscribers' lines which may only be able to provide a single or double speech channel and also on low traffic trunks which may be able to provide only 64K bit/sec. Hence the new technique is applicable at either a subscriber's terminal or at the first exchange in the network.

Figure 1 shows in highly simplified form such a system in which the circuitry is located at the subscriber's terminal. It is assumed that once set up, connections are over virtual circuits, and that the physical medium between the terminal and the packet switch is used in common by all sources at the terminal. A speech detector is provided which monitors the line to identify the presence or absence of speech.

The information to be conveyed, whether speech or data, is processed through a FIFO (first-in-first-out) store. Speech expressed digitally normally flows out of this store immediately after the header has been added, and continues to flow until a suitable pause in the speech occurs, or until the end of a packet length, e.g. 64 bytes, is reached, when a tail is added. If the speech talk-spurt continues without pause, a new packet, with its own header, is initiated.

In the case of data, the FIFO store is filled to the packet capacity, e.g. 1024 bytes, and then a channel, which provides the virtual circuit, is seized. Speech, because of the need to minimise delay, is handled "on the fly", so in a conventional system speech is only subjected to full packet delay when an output channel is unavailable or when the channel currently in use is of higher speed than the source. In the first of these cases, the congestion can be due to another speaker using the channel, or data using the channel.

Congestion may be tolerated if it causes relatively short delay at 64K bit/sec, i.e. from 0 to 8ms. However, delay due to data packet may cause an excessive delay if a speech packet has to wait. Hence when speech is detected while a data packet is being sent, a speech detector responds and terminates the data packet prematurely, providing it with a suitable tail. The speech packet is then provided with a header and transmitted, a tail being appended to it. Then the residue of the interrupted data packet is sent as a new packet, with its own

header and tail, which indicate that it is the residue of an interrupted packet. Thus the delay avoided lies, due to the length of a data packet, between 0 and 128ms, i.e. 64ms on average. Thus a serious source of delay is avoided at the originating end.

Figure 2 shows the structure of an X25 data packet used for data, while a speech packet as used in this system is similar with the exception that the FCS (i.e. error detection) check only applies to the header to ensure that delivery of the packet is effected. Note that any packet whose header does not check is rejected.

In Figure 2 the arrow indicates the direction of transmission. For a data packet the intelligence portion may contain up to 1024 bytes, while for speech it may also contain up to 1024 bytes. It will be seen that the header includes three portions, which in many cases would each consist of one byte, while the rest of the packet is in three portions. These three portions are a three-byte header, the information which may include up to 1024 bytes, and a three-byte tail. Propagation time may vary from nearly zero to as much as 500ms. Handling is assumed to include the delay at each stage, e.g. a maximum of 7ms per stage with 2ms average, plus 1ms for processing. For a packet of this size this gives a total delay here of 21ms. Thus the elimination of the delay to speech due to a data-occupied channel has a considerable effect on the overall delay.

Figure 3 is indicative of the application of the technique described above at the first switching centre. In the case of subscriber lines, the source of data ia apparent to the switching centre serving the lines, so the centre can exercise control if the incoming store thereat is not cleared when the data sending terminal is ready to send its next packet. If speech has to be conveyed in preference to data, the centre can terminate the current data packet as explained in connection with control at the subscriber's terminal. The centre has to be advised of the type of traffic being handled; this is conveniently done by using the control bit of a speech packet to give the speech identification to reserve the capacity.

An additional feature of the present arrangement relates to the avoidance of difficulties due to variable delay in the network, and is concerned with means by which the speech packets are time-stamped, i.e. they include an indication of the packets' time of origin. The method used to remove such variable delay is to accept a standard delay for each speech packet which is more than the delay normally expected in the network. Any speech packet whose delay exceeds the standard delay is discorded, while any packet which has less than the standard delay is caused to suffer this delay. This will often mean holding a speech packet in a buffer for the difference between the actual delay and the standard delay. This can readily be effected under clock and microprocessor control.

A modern telecommunication's network has to provide for satellite circuits which have inherently long propagation delays, while terrestial circuits have shorter delays. Hence the standard delay referred to above is not fixed, but is mode dependent on the delay suffered by the connection. This is automatically indicated by the difference between the time-stamp of a received packet and the time of reception thereof. Since the reassembly of the speech from the packets, with pause introductions where appropriate, is done at the receiving terminal, it is only the receiving terminal which needs to know the time-stamp. Therefore, it is transmitted with the packet, as mentioned above, and can conveniently be conveyed as the first part of the information. The degree of accuracy needed might be to 1ms, and would cover the period of one second, which should never be exceeded. Thus two bytes of the information can be allocated to the time-stamp, expressed duo-decimal or pure binary.

CLAIMS

1. An electrical intelligence transmission system, in which both speech and data are transmitted as packets over virtual connections set up in a fully digital network, the speech and data packets being transmitted at the bit rate of the transmission medium, in which speech packets have priority over data packets, in which, if a speech packet for transmission is detected during the transmission of a data packet, the transmission of that data packet is interrupted to allow the speech packet to be conveyed over the transmission medium, and in which when the medium again becomes free from speech, the non-transmitted part of the interrupted data packet is transmitted as a separate data packet.

2. A system as claimed in claim 1, in which each speech packet which is sent includes an indication of that packet's time or origin, in which at a receiving terminal the time of origin of a speech packet is compared with the time of reception of that packet, in which if the delay suffered by the packet exceeds a standard delay then that packet is discarded, and in which if the delay is less than the standard delay, the packet is buffered for a time such as to bring its delay to the standard delay.

3. A system as claimed in claim 2, and in which the standard delay is adjustable dependent on the characteristics of the transmission medium.

4. An electrical intelligence transmission system, substantially as described with reference to the accompanying drawings.

Printed in the UK for HMSO, D8818935, 3/86, 7102.
Published by The Patent Office, 25 Southampton Buildings, London, WC2A 1AY, from which copies may be obtained.